REMARKS

The FINAL Office Action (FOA) mailed May 9, 2009, has been carefully considered and these remarks are responsive thereto. Claims 1, 10 and 12 have been amended to correct "personal computers" to "a first personal computer" and "a second personal computer." Consequently, claims 4 and 15 are amended to read "first personal computer." Also, claims 10 and 21 have been amended to delete "entertainment" from "sound adapter." Claims 3 and 14 have been amended to clarify an antecedent issue with "program audio." Claims 5 and 16 have been amended to correct "sample" to "sampling." Claim 25 has been amended to add "a" before "sequence." New dependent claim 28 on independent claim 1 has been added to define the first and second sampling rates.

Applicants respectfully request reconsideration of pending claims 1-27 and new claim 28, address the several rejections made by the Examiner and answer each of his questions concerning the specification and support for the claims in the order presented by the Examiner in his FOA. A Request for Reconsideration is submitted herewith so that an Affidavit under Rule 132 may be entered in the record as support for a position taken in Applicants' earlier remarks and in this response that the specification, represented by, for example, US 2007/0189508 (the '508 publication, Knutson) published August 16, 2007, which is the publication of the present application and is used for reference purposes herein, fully enables the claims of the present invention, notwithstanding the Examiner's position to the contrary. See M.P.E.P 716.01 (A)(4).

Claim Rejections – 35 USC 112

Now turning to the **DETAILED ACTION** beginning at Page 2, pending claims 1-27 are rejected: "The claim(s) contain subject matter which was not described in the specification in such a way as to enable one skilled in the art to which it pertains, or with which it is most nearly connected, to make or use the invention." The claims are directed to <u>acoustic echo canceller</u> apparatus and a related method for use in one of a personal computer and a peripheral device.

It is respectfully submitted that one skilled in this art would have at least a bachelor of science degree in electronics engineering and experience in hardware and computer engineering as well as approximately two years of experience in the art. Mr. Benjuan Zhang meets or exceeds this level of skill as of March 5, 2004 and is assumed to be one reasonably skilled in the art. Mr. Zhang has been asked to review the present claim set, the Examiner's remarks regarding

lack of enablement, the present application in the form of the '508 publication (Knutson) and provide his remarks in the form of a Rule 132 Affidavit.

It is further submitted that the **Test of Enablement** is set forth in M.P.E.P 2164.01 as follows: "Any analysis of whether a particular claim is supported by the disclosure in an application requires a determination of whether that disclosure, when filed, contained sufficient information regarding the subject matter of the claims as to enable one skilled in the pertinent art to make and use the claimed invention. The standard for determining whether the specification meets the enablement requirement was cast in the Supreme Court decision of *Mineral Separation v. Hyde,* 242 U.S. 261, 270 (1916) which postured the question: is the experimentation needed to practice the invention undue or unreasonable? That standard is still the one to be applied. *In re Wands*, 858 F.2d 731, 737, 8 USPQ2d 1400, 1404 (Fed. Cir. 1988). Accordingly, even though the statute does not use the term "undue experimentation," it has been interpreted to require that the claimed invention be enabled so that any person skilled in the art can make and use the invention without undue experimentation. *In re Wands*, 858 F.2d at 737, 8 USPQ2d at 1404 (Fed. Cir. 1988). See also *United States v. Telectronics, Inc.,* 857 F.2d 778, 785, 8 USPQ2d 1217, 1223 (Fed. Cir. 1988) ("The test of enablement is whether one reasonably skilled in the art could make or use the invention from the disclosures in the patent coupled with information known in the art without undue experimentation."). A patent need not teach, and preferably omits, what is well known in the art," (our emphasis added). The so-called undue experimentation factors are listed in M.P.E.P. 2164.01(a) as comprising: (A) The breadth of the claims; (B) The nature of the invention; (C) The state of the prior art; (D) The level of one of ordinary skill; (E) The level of predictability in the art; (F) The amount of direction provided by the inventor; (G) The existence of working examples; and (H) The quantity of experimentation needed to make or use the invention based on the content of the disclosure.

M.P.E.P. 2164.01(a) requires: "It is improper to conclude that a disclosure is not enabling based on an analysis of only one of the above factors while ignoring one or more of the others. The Examiner's analysis must consider all the evidence related to each of these factors, and any conclusion of nonenablement must be based on the evidence as a whole." It is respectfully submitted that the Examiner has ignored Applicants' evidence of enablement and continues an enablement rejection without considering the evidence presented as a whole. Moreover, Applicants submit herewith evidence in the form of an Affidavit under Rule 132 that not only

tends to support Applicants' prior evidence but relates further evidence that the claimed embodiments and methods meet the **Test of Enablement**.

M.P.E.P 2164.04 clarifies: **Burden on the Examiner Under *>the< Enablement Requeirement [R-1]**. "Before any analysis of enablement can occur, it is necessary for the examiner to construe the claims. For terms that are not well-known in the art, or for terms that could have more than one meaning, it is necessary that the examiner select the definition that he/she intends to use when examining the application." To date, the Examiner has not specifically indicated a claim construction of the rejected claims 1-27. Later in M.P.E.P 2164.04, it is stated: "For example, doubt may arise about enablement because information is missing about one or more essential parts or relationships between parts which one skilled in the art could not develop without undue experimentation." In particular, the Examiner expresses doubt about sampling rates (signal conversion), delay matching and average processor load as will be further discussed herein.

It is respectfully submitted that the Examiner has failed to make a *prima facie* case of lack of enablement. On the other hand, M.P.E.P 2164.05 further clarifies "To overcome a *prima facie* case of lack of enablement, applicant must demonstrate by argument and/or evidence that the disclosure, as filed, would have enabled the claimed invention for one skilled in the art at the time of filing," that is as of the date of the filing of the present application under the PCT or March 5, 2004. Moreover, M.P.E.P 2164.05 warns "The examiner should **never** make the determination based on personal opinion. The determination should always be based on the weight of all the evidence."

Firstly, the Examiner has rejected all claims 1-27 as failing to meet the enablement test (without ever referring to the test of "undue experimentation.") Indeed, it is not clear whether the Examiner considered the test (2164.01) at all. Deficiencies in the specification and claims as originally filed appear to lie in three areas: a concern expressed by the Examiner that no sampling rate conversion (digital to analog or analog to digital conversion) is specifically discussed regarding sample rate conversion, no delay line matching calculation is provided and that one skilled in the art would not be able to determine an average load of a processor. These deficiencies as well as related alleged deficiencies are discussed in the Rule 132 affidavit and below.

Claims 10, 11 and 21, 22 do not recite any of the complained of elements and so are enabled

On the other hand, the Examiner has failed to discuss any element of any of claims 10-11 and 21-22 which is not enabled. The Examiner has not specifically identified any feature of any of these claims that is allegedly not enabled. The Examiner, having failed to identify any feature not enabled by these claims, is respectfully requested to withdraw his rejection of claims 10-11 and 21-22 for failure to meet the enablement test. Moreover, claims 10 and 21 have been amended to delete "entertainment" thus alleviating any concern about the meaning of that term.

Alleged lack of enablement of remaining claims

Repeatedly, Applicants have pointed to specific elements or relationships between elements that are fully described by the specification, and the Examiner has ignored Applicants' guidance and provided in its place his own opinion. One of many examples is the Examiner's statement: "Applicant's specification does not disclose any means or relationships between sampling rate of the incoming microphone signal (telecommunications signal) and the 'higher sampling rate' of the non-training audio signal. Applicants do not show a sampling stage in any of the admitted figures." Yet, Applicants have repeatedly rebutted this remark. Applicants have pointed, for example, to Figure 5 and paragraph [44]: "The first sample rate converter 536 and the second sample rate converter 546 are utilized to perform sample rate conversion to match the entertainment sample rates (typically 44.1 or 48 Ksps) to the communication sample rate (typically 8 Ksps)." One skilled in the art would recognize the 8 Ksps sample rate as consistent with the later described analog telecommunications bandwidth: "The audio communication bandwidth corresponds to audio conferencing and telephony and may be, but is not limited to, 300 Hz to 3.3 KHz. Echo cancelling over this bandwidth will save processing power." Thus, it has been and continues to be respectfully submitted that sampling rates, sampling "stage" and the like are, for example, shown by FIG.'s 1 and 5 and its related description as inherent in the depicted analog microphone and speaker made digital via analog signal conversion apparatus that would be well known to one of ordinary skill in the art and would be known to be included without undue experimentation from the Knutson description. As will be further discussed herein, the embodiment of FIG. 8 and its related description is also supportive of the claims.

Applicants will now address each comment or question made by the Examiner at section 2), Pages 2-4.

The Examiner continues to state at Pages 2-3, 2.: "The claims recite an entertainment (non-training) audio signal sampled at a higher sampling rate than the first sampling rate of the

input microphone. Applicants' specification does not disclose any means or relationships between sampling rate of the incoming microphone signal (telecommunications signal) and the 'higher sampling rate' of the non-training audio signal. Applicants do not show a sampling stage in any of the submitted figures. Additionally, it is not clear how the 'non-training' signal is presented to the user if it is not converted to an analog signal to be driven out of the speaker (in order to perform the claimed 'training function'). Applicant's specification does not specify if/when/where the signal is converted to analog. Is applicant claiming an 'entertainment audio signal' that is transmitted through the speaker in digital form? That does not make sense as that would not be a very 'entertaining' signal." This quotation will be responded to first as further comments/rejections of section 2. are related to different subject matter. The quotation is again broken down into the following: <u>Sampling rate</u> and <u>Digital to Analog/Analog to Digital</u> where one skilled in the art would readily recognize that a speaker should receive analog and analog from a microphone would be converted to digital for use in a related device such as a personal computer, a related device to a personal computer or a peripheral. The related nonsensical transmission of digital through an analog speaker or reception of digital through a microphone is an example of their inclusion without undue experimentation by one of ordinary skill in the art.

<u>Sampling Rate</u>

Claims 1, 5, 12, 16 and 23 use the term "sampling rate." The term "sampling rate" is not construed by the Examiner but is clearly described in the specification as discussed below.

It is respectfully submitted that the present application describes an Acoustic Echo Canceller with Multimedia Training Signal. Reference will be made throughout to US 2007/0189508 (Knutson) published August 16, 2007, which is a publication of the present application under examination. Acoustic echo cancellation, according to the BACKGROUND OF THE INVENTION paragraph [0004], relates to acoustic echo defined as follows: "An acoustic echo is an undesirable condition that results from sound that emanates from a speaker being fed back into a microphone. To reduce or eliminate such echo an acoustic echo canceller is employed." Referring briefly to FIG. 3A-3C, acoustic echo paths (AEP) are shown for a personal computer 399, a mobile computer 398 and a PDA 397. The Examiner does not appear to disagree that the specification relates to an acoustic echo canceller and method. On the other hand, the Examiner relies on other echo problems than <u>acoustic echo</u> such as sidetone and double-talk.

A problem for an <u>acoustic</u> echo canceller as defined is one of training [0005]. It is admitted that one approach to acoustic echo cancellation involves known adaptive filters. Given the environment in which, for example, a personal computer or peripheral device may find itself such as a personal computer/peripheral device in a Starbucks coffee house, it is described that the coefficients and training of the adaptive filter are a problem.

In the SUMMARY OF THE INVENTION section, paragraph [0008], it is described: "The present invention is provided on a device capable of conferencing (including videoconferencing and teleconferencing) and/or telephony (including Internet Protocol (IP) telephony) applications as well as audio applications, such that the audio applications are used to train the echo canceller in the background." The depicted sound adapter (sound card) 199 and buffers 172 in the sound adapter are typically provided for implementing audio applications and, in embodiments and methods, are used to train an acoustic echo canceller.

The Examiner is now referenced, for example, to FIG. 1, and elements CPU 102, ROM 106, RAM 108 including software buffer 171, and a loop shown connected to these via bus 104 comprising Sound Adapter 199 with hardware buffer 172 [0034] (an alternate or in addition to software buffer 171), microphone 151, (acoustic) echo canceller 152, speaker 150 where connections are shown between them and are further described by the specification. Of these, it would be clear to one skilled in the art from FIG. 1 and FIG. 5 without the use of undue experimentation that bus 104 is digital in nature and devices connected thereto must operate digitally. Per paragraph [0029], "The electronic device 100 includes at least one processor (CPU) 102 operatively connected to a system bus 104. A read only memory (ROM) 106, a random access memory (RAM) 108, a display adapter 110, an I/O adapter 112, a user interface adapter 114, <u>a sound adapter (also referred to herein as an 'audio card' 199</u>, and a network adapter 198, are operatively coupled to the system bus 104," (our emphasis added) which one skilled in the art would understand means digital coupling. Moreover, it is clear to one of ordinary skill in the art from FIG. 1, FIG. 5 and FIG. 8 that, for example, a microphone 151, 512, 812 and a speaker 150, 514, 814 are analog in nature and, for example, a sample rate conversion via depicted elements 536, 830 and 546, 852 requires analog to digital or digital to analog conversion at a speaker or microphone without undue experimentation to recognize such.

The Examiner is further referred to FIG. 5, described at paragraph [0017], as a block diagram illustrating an <u>acoustic</u> echo canceller 400 to which the present invention may be

applied, according to yet another illustrative embodiment of the present invention. One skilled in the art would understand from FIG. 5 and the description thereof in the context of the present specification and the rest of the drawings that, for example, an entertainment signal is audio output 597 of sound adapter 199 in digital form [0042]. "The acoustic echo canceller 500 is capable of operation at different audio sample rates." First 536 and second 546 sample rate converters are then introduced.

Paragraph [0044] then begins: "The first sample rate converter 536 and the second sample rate converter 546 are utilized to perform sample rate conversion to match the entertainment sample rates (typically 44.1 or 48 Ksps) to the communication sample rate (typically 8 Ksps)." Thus, notwithstanding the Examiner's comments to the contrary, digital sample (sampling) rates are clearly spelled out in the specification. One skilled in the art would recognize the 8 Ksps sample rate as consistent with the later described analog bandwidth. Per paragraph [0044], "The audio communication bandwidth corresponds to audio conferencing and telephony and may be, but is not limited to, 300 Hz to 3.3 KHz. Echo cancelling over this bandwidth will save processing power." Also, it is clear that there is a relationship between sampling rates where, for example, according to claim 1, "said entertainment sound adapter output being sampled at a second higher sampling rate than said first sampling rate." Claim 28 now defines first and second sampling rates according to the specification in the event that claim 28 may overcome the Examiner's concern.

Thus, it is again respectfully submitted that digital sampling rates, analog signal conversion and the like are, for example, shown by FIG. 5 and its related description without undue experimentation. Again, acoustic echo cancellation relates, for example, to adaptive filter 516.

Now, the sampling associated with FIG. 8 will be discussed. Note that audio sources 898 output a "stream" toward speaker 814; otherwise, windows stream buffer 824 would make no sense. Consequently, one skilled in the art would recognize the stream is digital and must be converted to analog at buffer 822 output or speaker 814 input. Similarly, microphone 812 provides an input to a hardware input buffer 840 (which is digital at microphone output or buffer input) which after recording control interface 844 is provided to windows stream buffer 846 (clearly a digital device). Consequently, one skilled in the art would readily recognize that sampling occurs at 822/814 or 812/840 and conversion occurs at 536, 546, 830, 852 without

undue experimentation when all the enablement factors and evidence are given proper consideration. Reconsideration and withdrawal of the rejection of claims based on non-enablement of "sampling rate" or its conversion is respectfully requested.

Digital to Analog/Analog to Digital

One of the factors regarding enablement is the scope of the claims. The claims do not require analog to digital or digital to analog conversion. Consequently, no analog to digital conversion needs to be shown in the figures or discussed in the specification. More importantly, it would be recognized by one of ordinary skill in the art that analog signal conversion would be utilized at analog speakers and microphones in a digital system without having to resort to undue experimentation to provide the conversion. The speaker 150, 514, 814, clearly an analog speaker, receives digital output of known sound cards, audio cards or sound adapters and outputs the same via provided but a not specifically shown digital to analog converter which may comprise part of the speaker or may be associated with the speaker 150, 514, 814 as one skilled in the art would readily recognize or at 822/814 without undue experimentation. Similarly, microphone 151, 512, 812 receives a sound signal from an external source (such as a person on a telephone call), the microphone 151, 512, 812 acts as a transducer to convert sound waves to an analog electrical signal and, as one skilled in the art would understand without undue experimentation, may be converted by an analog to digital converter that is a part of the microphone electronics 151, 512, 812 or separate from it (not shown) for digital audio input 598 to computer 599 or at 812/840. In other words, the Examiner is correct that A/D or D/A conversion would be inferred by one skilled in the art at microphone 150, 512, 812 and speaker 150, 514, 814 respectively. It would be nonsensical to think otherwise.

Finally, the Examiner states in conclusion at Page 3: "For the purpose of examination, the Examiner assumes the claims are referring to the inherent step of matching the sampling rates of the training-signal (entertainment signal) with the signal-to-be-modified-via-the-training signal (the telecommunications signal input from the microphone)) for the inherent purpose of aligning up the samples so that accurate adaptations can be made." Aside from the use of the word "inherent" and the purpose of aligning up the samples with which Applicants strenuously disagree, the Examiner appears to have come to a correct assumption with respect to sampling rates and matching them without undue experimentation.

But, as quoted from the FOA, Applicants cannot agree with "inherent" because Applicants have stated that an advantage of such sampling and matching is to "save processing power" [0044]. The Examiner addresses this power saving advantage at page 14 as follows: "The 'power savings' mentioned by applicant is the fact that one high rate signal is resampled to a lower rate which allows the adaptive echo canceller to function at a lower clock rate, which saves power . . . the echo canceller must receive two signal [sic] at the same sampling rate in order to function correctly. Applicant's claimed advantage of matching sampling rates is an inherent feature of any digital echo canceller in a bidirectional system." Yet, the Examiner is stating personal opinion that two sampling rates require resampling or aligning which is opinion and contrary to the specification. The Examiner's statement is a good example of the use of improper hindsight to judge an advantage of sampling rate matching.

Applicants also wish to raise the significant point that the Examiner relies upon US Patent No. 5,553,137 (Nyhart et al) to reject the present claims. Yet, the Examiner admits that Nyhart is digital and its microphone 106 must inherently have analog to digital conversion (not shown in Nyhart's drawings or discussed in Nyhart's application). The Examiner takes the position that it is okay for Nyhart not to show analog to digital conversion, it is inherent. Yet this same Examiner's position results in non-enablement for applicants: at Page 12, for example, the Examiner states: "Nyhart . . . inherently comprises an analog-to-digital converter, which will sample the training audio in the same microphone input that receives the telephone signaling (for a conferencing application for example). The ADC inherently comprises a 'sample rate converter' which will resample any input signal into the preset sampling rate (which will be the same as the telephone signaling (conferencing application)." Applicants consequently traverse the Examiner's non-enablement rejection on the grounds that even Nyhart, relied upon by the Examiner, fails to show an ADC for its microphone and yet Nyhart is presumed to have issued with a presumption of validity (absent an ADC). One skilled in the art would readily appreciate that ADC or DAC is required at an analog microphone or speaker as the Examiner has readily appreciated to be the case in Nyhart for sample rate conversion to occur without undue experimentation when all the enablement factors and evidence are given proper consideration.

Now Applicants will respond to the Examiner's rejection of the claims related to delay matching buffers.

Delay Matching Buffers

Delay matching is a feature of claims 6 and 17: "a delay matching buffer for matching a delay of the first path with a delay of the second path." There is ample support for this matching in the present specification.

Beginning at the middle of Page 3 of the NFOA, the Examiner states: "the Examiner makes an enablement rejection to applicants' claims that recite the delay matching buffers as applicants' specification does not provide enough information for one skilled in the art to ascertain the desired delay and further how to compensate for the delay." Although not indicated by the Examiner, only claims 6/1 and 17/12 appear to relate to this enablement rejection. The Examiner again states his personal opinion that there must be some "synchronize" activity of the software buffer 171 or the hardware buffer 172. Such is not the case, "synchronize" cannot be found anywhere in the Knutson et al. specification because it is not necessary.

Consequently, it is appropriate for the Applicants to address the application of buffers 171 and 172 as described in the specification and shown in the drawings and as one skilled in the art would understand from the specification and drawings.

A first mention of buffers in found at [0034] with reference to FIG. 1: "The electronic device 100 further includes buffers 171 that are included in RAM 108 and also buffers 172 that are included in the sound adapter 199." One skilled in the art would understand that buffer 171 may be considered a software buffer and 172 a hardware-related buffer.

As already indicated above, FIG. 5 represents one embodiment of an acoustic echo canceller of the claims. As described at paragraph [0042], "FIG. 5 is a block diagram illustrating an acoustic echo canceller 500 to which the present invention may be applied, according to yet another illustrative embodiment of the present invention. The acoustic echo canceller 500 is inserted between the speaker 514 and microphone 512 of a personal computer 599 having and audio input 598 and an audio output 597. The acoustic echo canceller 500 is capable of operation with different audio sample rates. The acoustic echo canceller 500 includes an adaptive filter 516, an adder 518, a multiplier 520, a <u>first delay matching buffer 532</u>, a first Low Pass Filter (LPF) 534, a first sample rate converter 536, <u>a second delay matching buffer 542</u>, a second Low Pass Filter (LPF) 544, and a second sample rate converter 546" (our emphasis added).

It is then indicated at [0045], "The first delay matching buffer 532 and the second delay matching buffer 542 are utilized to match buffer delays <u>at the different sample rates.</u> The buffers

that are to be matched by the first delay matching buffer 532 and the second delay matching buffer 542 may be, for example, software buffers (e.g., buffers 171) and/or hardware buffers (e.g., 172) as described herein" (our emphasis added). It is respectfully submitted that one skilled in the art would be able to utilize the delay matching buffers for the described purpose. However, there is another use for the buffers in connection with other buffers as described in the specification.

At paragraph [0048]: "Since in MICROSOFT WINDOWS and other non-real-time operating systems audio is implemented by buffering data to the speakers and from the microphone, the system will operate in bursts, as instructed by the operating system, processing buffers full of data. Filter coefficient adaptation would proceed as described and illustrated with respect to FIG. 6" which relates to a flow diagram for managing processor load in relation to the adaptive filter of the various embodiments.

Then, beginning at [0049], the embodiment of FIG. 8 is discussed at [0050]-[0051]: "A playback volume control user interface 820 is capable of controlling the playback volumes of the plurality of audio sources 898. It is to be appreciated that any stream buffer delays induced prior to the plurality of audio sources 898 being input to the playback volume control user interface 820 do not apply to the echo cancellation problem. The playback volume control user interface 820 is coupled to a hardware output buffer 822 and to a WINDOWS stream buffer 824. The hardware output buffer 822 is also coupled to the speaker 814. The WINDOWS stream buffer 824 is coupled to an output delay matching buffer 826 that, in turn, is coupled to a Low Pass Filter (LPF) 828. The LPF 828 is coupled to a sample rate conversion device 830 that, in turn, is coupled to an adaptive filter 832. The adaptive filter 832 is coupled to a multiplier 834 and an adder 836. The multiplier 834 is also coupled directly to the adder 836."

Per paragraph [0051], "The microphone 812 is coupled to a hardware input buffer 840 that, in turn, is coupled to a recording control user interface 844. The recording control user interface 844 is coupled to a WINDOWS stream buffer 846 that, in turn, is coupled to an input delay matching buffer 848. The input delay matching buffer 848 is coupled to a Low Pass Filter (LPF) 850 that, in turn, is coupled to a sample rate conversion device 852. The sample rate conversion device 852 is also coupled to the adder 826."

The connections are explained further in paragraphs [0054] – [0055] with reference to FIG. 6 and FIG. 8 as follows: "It is presumed, but not necessary, that the system to which the

acoustic echo canceller 800 is to be applied includes a sound card. In the case that the present invention is applied to a system having a sound card, buffers (e.g., WINDOWS stream buffers 824 and 846) are used to couple streams of samples to and from the sound card. The buffers in this case are software structures (e.g., such as buffer 171 shown in FIG. 1) that store enough samples so that WINDOWS applications can fill/empty a buffer without the buffer running out or overfilling between OS task switches. Also, there are hardware buffers on the sound card (e.g., such as buffer 172 shown in FIG. 1) for audio playback or capture (from the microphone). The buffer delays can be significant in the WINDOWS environment. Thus, the delay of the adaptive filter 832 needs to be adjusted <u>to span the acoustic echo delay range</u>, without the need for incorporating the buffer delays in the delay span of the adaptive filter 832. To handle up to 100 ms of echo, an absolute minimum of 800 taps are needed at 8 Ksps. More taps would be provided (e.g., 1024 taps) so that each echo can be a filter to match the phase, amplitude, and general frequency response of <u>each echo path</u>.

In a system where the WINDOWS buffers (e.g., such as buffer 171 shown in FIG. 1) and the hardware input buffers and hardware output buffers (e.g., such as buffer 172 shown in FIG. 1) are identical, then the delay matching buffers (e.g., buffers 826 and 846) would be non-existent. However, per [0055], "the delaying matching buffers 826 and 846 are included in FIG. 8 <u>so that the path from the speaker 814 back to the adaptive filter 832 has the same delay as the path from the microphone 812 back to the adaptive filter 832</u>" (our emphasis added). This is a clear description of delay matching per the rejected claims requiring no undue experimentation. Consequently, further advantages/uses of the delay buffers are for adaptive filter control and microphone/speaker delay to adaptive filter compensation. One skilled in the art would understand the latter is calculable from the predetermined delays of the described paths without undue experimentation. Moreover, the latter may vary from computer to computer 499, 599, 799 or in FIG. 8 (where the "personal computer" may have "a plurality of audio sources 898" [0049].

Further description of buffers found at paragraphs [0060]-[0061] where one skilled in the art would understand a software buffer 171 is described whereby processor load reduction can be achieved by use of buffers: "Adaptation and filtering are only completed during every adapt call to the method/routine of FIG. 6. Otherwise, data is only stored in the filter input buffers (but adaptation is not performed), reducing the computational load from 2n operations per sample to one operation per sample, where n is the number of taps in the filter, given a full-band LMS echo

canceller. Thus, it is determined whether the value of the adaptive counter is greater than or equal to a pre-specified adaptive counter comparison value (step 620). If so, then the value of the adaptive counter is reset (to zero) (step 625), and the method proceeds to step 630. Otherwise the input buffers and the adaptive filter buffers are updated, but the adaptive filter is not operated (step 655), and a return is made to the operating system." Further description is provided at paragraphs [0064] and [0068] but the essence of the utilization of the delay matching buffers has already been discussed.

Thus, in summary, Applicants are not "synchronizing" as assumed by the Examiner. Rather, one skilled in the art would know to utilize delay matching buffers in the manner indicated by the specification and as broadly indicated by claims 6 and 17 "for matching the delay of the first path with a delay of the second path." Consequently, Applicants respectfully request that, when the test for enablement is applied to claims 6 and 17, the specification and drawings pass the test of "without undue experimentation." One skilled in the art would readily appreciate how to implement the claimed feature of a delay matching buffer.

At Page 4, first complete paragraph of the FOA, it is asserted with respect to claims 1, 6, 7, 8 and 27 that "applicant has not provided any specific details as to how the system would monitor the total load on the processor (or even every process handled by the processor). Further, the applicant has not [providing] (provided intended) any timing diagrams or algorithms by which the processor determines the 'average load.'" Any need for timing diagrams is a personal opinion again of the Examiner and not that of Applicants. Applicants will now address alleged non-enablement of <u>Processor Load</u>.

<u>Processor Load</u>

Processor load appears to be related to the following claims: 7-9, 18-19 and 27. These claims relate to "an average load threshold" or "average load." The Examiner raises the question what is an average load and the answer depends on the application of the processor and the type of operating system.

One skilled in the art of computers and operating systems in which the present acoustic echo canceller is employed would readily recognize that the operating system may provide such output as indications of load. As described, for example, in paragraph [0058]-[0059]: "Adaptation and filtering are only practical when audio is coming out of the system. Either an audio application must be running, or the Operating System (OS) must generate a sound. Thus, it

is determined whether an audio application is currently being executed by the OS or whether the playing of a sound is being initiated by the OS (step 605). If so, the method proceeds to step 610. Otherwise, a return is made to the operating system. It is to be appreciated that the sound may be, but is not limited to, a sound relating the arrival of an e-mail, an indication sound of some event (e.g., a notification of an incoming call, a conference call reminder, a warning, etc.), and even a pre-specified sound sequence also used for a purpose other than solely training the echo canceller." It is respectfully submitted that the Examiner's suggestion of timing diagrams and the like to show processor determination of average load is inapplicable. To the contrary, FIG. 6 and its attendant description is urged to provide thorough details of a processor load algorithm which does not require undue experimentation. The specification continues in [0060] already quoted above, and [0061]-[0065]. It is respectfully submitted that the specification and FIG. 6 provide ample discussion of processor load and the use of an adaptive counter for acoustic echo canceller control.

As supported in the attached Affidavit under Rule 132, a typical average load of a personal computer is readily available through TOP or WINDOWS provided task management to output an average processing load. A threshold range for a typical computer is well known to one skilled in the art for a non-real time computer but may vary from computer to computer and whether the computer must operate in real time or not. Thus, when the load is lower than an average, training can occur, and the echo canceller is run continuously using all audio samples at step 650; otherwise, if high, the adaptive counter is incremented at 615. See, for example, paragraph [0059].

The Examiner comments on claims 6-8 and 27 that average load and processor load are unclear. The Examiner is referred to the attached affidavit, specification, drawings and the discussion above of these terms. It is respectfully submitted that one skilled in the art would understand these terms from the specification (for example, paragraphs [0058] and [0059]). Per paragraph [0059], for example, "Depending on the average processor load, different approaches can be taken to adapt the acoustic echo canceller filter. Thus, at step 610, it is determined whether the average processor load is low or high. If the average processor load is low, then the acoustic echo canceller can operate continuously, using all audio samples (step 650), and then a return is made to the operating system. Otherwise, if the average processor load is high, then the filter is adapted intermittently. To adapt the filter intermittently, a counter

(hereinafter "adaptive counter") is used, and a value of the adaptive counter is incremented (step 615). It is to be appreciated that the present invention is not limited to the use of a counter to intermittently adapt the adaptive filter and, thus, other approaches may also be employed while maintaining the spirit of the present invention."

When the totality of the evidence is considered and in view of FIG. 6 and its detailed description and the fact that one skilled in the art would readily recognize an average load for a given processor and operating system, Applicants respectfully submit that the specification enables one to practice the claims and "average processor load" without undue experimentation.

The Examiner then introduces a "double-talk detector" at Page 4 as providing a well-known function of adaptive filter/audio training without citing to a particular reference. Applicants will now briefly address a Double-talk Detector.

Double-talk Detector

To the extent understood by Applicants, double-talk detection relates to near end/far end signal detection. The acoustic echo canceller described by the Knutson et al. specification is directed at the training of an acoustic echo canceller per FIG. 1, FIG. 5, or FIG. 8, for example, where, for example, computer 599 is or may be involved in, for example, internet or IP telephony. A double-talk detector suspends or reduces echo canceller adaptation in the presence of bi-directional voice communications. To the contrary, the present application is directed to background training of an acoustic echo canceller. The double talk path is a different path than AEP of FIG. 3A, 3B and 3C. Double talk detection may be added to an IP telephone PC system, but a double talk detector appears to teach nothing that assists in background training of an acoustic echo canceller. The Examiner is requested to explain his position further with some citation to a reference so that Applicants may better understand the Examiner's comments. However, without a specific citation to an embodiment of a double-talk detector that relates to an acoustic echo canceller having background training using an entertainment signal audio output 597 from a sound adapter, audio card, sound card 199, Applicants cannot effectively respond to the Examiner's assertion.

Entertainment

At Page 5, Section 2, the Examiner rejects claims 1-27 as indefinite for use of the phrase "entertainment." Applicant has deleted "entertainment" from claims 10-11 and 21-22 and the rejection of these claims is believed to be rendered moot thereby. The Examiner alleges that the

phrase is not clearly defined. Yet, entertainment is defined by way of example, for example, at [0024], as: "entertainment (e.g. music, multimedia, etc.)." The Examiner calls the definition of entertainment "vague" and fails to construe the term. To the contrary, the Examiner goes beyond the specification and suggests that tones and noise fall within "entertainment." Thus, Applicants continue to traverse the alleged vagueness of the term "entertainment."

At Knutson paragraph [0044], it is stated: "at high sample rates for entertainment quality audio" where it is recognized that audio music, for example, may involve a high sampling rate in order to enjoy if played as an analog signal through a speaker.

There is provided a grocery list of what may be considered entertainment that falls within this definition at Knutson paragraph [0047]: "video games, playing MP3s, CDs, or other audio files, playing video files." Finally, at paragraph [0064], it is stated: "What the microphone hears is not being used in communications when entertainment applications are running." In summary, the Examiner is referred to the following paragraphs for entertainment, application and sound adapter: [0024], [0029], [0044], [0047], [0054] and [0064]. It is respectfully submitted that one skilled in the art would readily appreciate the meaning of "entertainment sound adapter" and "entertainment application" by way of the examples provided in the specification of "entertainment," entertainment applications and a sound card, audio card, sound adapter used for "entertainment."

Claim Rejections – 35 USC 103

The Examiner at Pages 6-12 provides a detailed rejection based on Nyhart et al., US 5,553,137 including an **Additional set of rejections**. The Examiner responds to Applicants' arguments at pages 13-20 which Response, for the most part, contains prohibited "personal opinion" pursuant to M.P.E.P 2164.5 which warns: "The examiner should **never** make the determination based on personal opinion. The determination should always be based on the weight of all the evidence." The Examiner at pages 13-17 seems to argue both lack of enablement and at the same time inherency using Nyhart. At page 17, first paragraph, the Examiner states: "The examiner notes that applicant's specification gives no implementation details as to the specific algorithms used, and the spec provides nothing more than block diagrams to show the claimed invention. The examiner contends that if it truly is not well known how the adaptive filters/synchronization of a digital echo cancellation are implemented, then applicant's specification and claims will be subject to additional 112 first paragraph enablement

rejections." As is well known to one of skill in the art, "synchronization" is not a requirement of simply stepping down a single computer clock reference of a computer for sampling since the sampling frequency comes from a single clock source.

Applicants will address the claimed acoustic echo cancellation out of sequence of the Examiner's stated rejections because Applicants believe that understanding Nyhart and the sidetone echo Nyhart is canceling versus the disclosure and claims of the present invention is key to an appreciation of the differences between Nyhart et al.'s sidetone echo canceller and the claimed structure of claims 1-27 and new claim 28.

Claim 1 is repeated here for clarification: A method comprising:

implementing a telecommunications application of an electronic device, said electronic device comprising <u>one of a first personal computer and a peripheral device for use with a second personal computer</u>;

<u>sampling a telecommunications signal of said telecommunications application at a first sampling rate</u>; and

utilizing sound output of an entertainment sound adapter of said electronic device, <u>said entertainment sound adapter output being sampled at a second higher sampling rate than said first sampling rate</u>, said entertainment sound adapter output corresponding to <u>a non-training audio application of said electronic device to train an acoustic echo canceller in a background of said telecommunications application</u>.

The method of claim 1 is clearly described and supported as discussed above, for example, by FIG.'s 1 through 8 and the specification when taken as a whole. Acoustic echo cancellation is defined as the echo resulting from "sound that emanates from a speaker being fed back into a microphone." The Examiner is referred to the acoustic echo paths (AEP) shown in FIG. 3 which occur in the air as feedback from speaker to microphone and not entirely within a telephone system. Nyhart et al. relates to a different type of echo – sidetone echo. Beginning at col. 1, line 13, Nyhart states: "Many telephone systems, particularly cordless telephone systems, are defined to operate in urban environments which have a high level of ambient noise. In telephony, sidetone is defined as an attenuated level of one's own voice heard in the telephone handset ear piece. In telephony systems in which there is a delay to the audio path, the sidetone produced by the 4 wire to 2 wire reflection (from a conversion hybrid such as "standard hybrid" 128) will sound like echo. This (sidetone) echo can be annoying to the user to the point of

disrupting the ability to communicate on the telephone." Sidetone occurs in the handset (not in the air) among the microphone, hybrid and speaker so that a person's voice can be heard at the speaker, and the echo is caused by the hybrid 128. See also col. 2, ll. 63-66. Hence, per Nyhart FIG. 1, the Nyhart echo cancellation of sidetone echo involves the base station 102 and the controller 122, DSP 124 and hybrid 128 with paths 130. An air path AEP between speaker 108 and microphone 106 has nothing to do with sidetone echo cancellation. An acoustic echo path is shown in Knutson FIG. 3A, 3B and 3C "relating to the feedback of a speaker output to a microphone input" of the devices shown. The Examiner fails to appreciate the difference between sidetone echo cancellation and acoustic echo cancellation. The Examiner fails to appreciate the difference between an electronic device such as a personal computer and a peripheral and a cordless telephone. The Examiner fails to appreciate the significance of sampling at first and second sampling rates where, for example, entertainment audio is sampled at a high rate and telephony at a low rate and matched at the lower rate as indicated above for the advantages indicated above when Nyhart only relates to telephony. Moreover, the Examiner fails to appreciate: "said entertainment sound adapter output corresponding to <u>a non-training audio application of said electronic device to train an acoustic echo canceller in a background of said telecommunications application</u>."

Nyhart only mentions microphone 106 and speaker 108 once in Nyhart's entire specification.

Until the Examiner is able to understand that there is a clear difference between an <u>acoustic</u> echo canceller and associated structure and a <u>sidetone</u> echo canceller and related structure, Applicants continue to maintain their position that claims 1-27 (and new claim 28) are patentably unobvious in view of Nyhart or applicants' admitted art applied thus far by the Examiner.

Applicants respectfully request an interview with the Examiner to discuss independent claims 1, 10, 12, 21 and 23 and must respectfully repeat the same arguments and remarks made previously in response to the plurality of office actions issued in this application. Applicants respectfully submit that a dialog with the Examiner may result in an understanding by both of each other's respective positions and perhaps to passage to issue of the present application.

The Examiner in the most recent FOA rejects claims 1-4, 9-15, 20-22 and 27 under 35 USC 103(a) as being anticipated by Nyhart et al. (5,553,137). The Examiner further rejects

claims 5, 7-8, 16, 18-19 and 23-26 under 35 USC 103 as being unpatentable over Nyhart et al. (5,553,137) (hereinafter, Nyhart) as applied to claims 1 and 12. The Examiner rejects claims 6 and 17 under 35 USC 103 as being unpatentable over Nyhart as applied to claims 1 and 12 and further in view of applicant's admitted prior art (spec).

The Examiner is off base on all counts. For example, the Examiner states at Page 9, 4. regarding claims 6 and 17, "Applicant's admitted prior art discloses well known adaptive filters used to perform the echo cancelling. The digital system <u>inherently</u> comprises means to delay all signal paths so as to synchronize the signals (to give 'real time' bidirectional communication.) (spec pages 1 and 2). It would have been obvious to one of ordinary skill in the art at the time of this application to implement well known echo canceller features like a filter and delay means for the purpose of implementing the disclosed canceller" (our emphasis added). It cannot be stated enough that Nyhart and Knutson et al. provide different echo cancellers in different structures and their methods are for different purposes. Knutson teaches no need for synchronization. The Examiner's inherency arguments all fail as personal opinion without basis in fact for impermissible hindsight reconstruction. Moreover, Nyhart, for example, describes no delay, no entertainment application or any need to match delays or discussion of processor load.

Again, the "admitted prior art" at paragraph [0005] of the specification comprises a reference to adaptive filters used to perform acoustic echo cancelling. Indeed, an "adaptive filter" is disclosed at paragraph [0005], where it is further stated that "the stored coefficients will be invalid or possibly worse than starting from a zero coefficient point." At paragraph [0006], Applicants state: "Another approach involves reducing the local speaker volume when a local user is speaking into the microphone so as to reduce the canceling requirements of the adaptive filter." Nyhart fails to address these problems, barely mentioning a microphone and a speaker of a cordless phone.

Consequently, the Examiner's use of so-called admitted prior art is respectfully traversed as teaching anything other than what is stated in the specification. The Examiner must provide some citation to a reference which teaches what the Examiner alleges the prior art teaches beyond the admitted prior art at pages 1 and 2 of the specification, for example, "The digital system inherently comprises means to delay all signal paths so as to synchronize the signals. . ." This is a statement of opinion, not fact, and certainly does not come from Nyhart.

With respect to "inherently comprises means to delay," the Examiner states at page 9 of the FOA: "It would have been obvious to one of ordinary skill in the art at the time of this application to implement well known echo canceller features like a filter and delay means for the purpose of implementing the disclosed canceller." This does not come from Nyhart and is a statement of opinion, not fact. Then, at page 12, the Examiner repeats the same comments as part of an **Additional set of rejections.**

This is but one example of the confusion as to what is rejected and on what basis when the burden is on the Examiner to demonstrate a *prima facie* rejection based on obviousness and non-enablement.

<u>Features of claimed embodiments missing from Nyhart</u>

We now again summarize the allegations of the Examiner regarding Nyhart and identify features of claimed embodiments missing from Nyhart. Nyhart allegedly teaches non-training audio according to the BACKGROUND (FOA, Page 10, 6.) regarding claims 1-4, 9-15 20-22 and 27. The Examiner states that Nyhart trains on non-training audio referring to the incoming microphone signal regarding claims 1 and 12. Reference is made to Col. 4, lines 20-30 for support. Yet this reference is to DECT – Digital Enhanced Cordless Telecommunications – which is not a reference to non-training audio. Further claims 2, 13, then 3, 11 14, 22 with a comment "it may be audio" are also discussed with no specific reference to any support in Nyhart.

There is a discussion in the Nyhart BACKGROUND of using "sidetone" which is the intentional combination of microphone pick-up to be heard by the near end caller or background noise which can result in the following: "an increased chance that the near end user will begin speaking before convergence. This in turn results in the near end user initially hearing his sidetone as the canceller converges. If the noise level is increased to a level higher than background noise, the far end user may hear the added noise for the duration of the training of the canceller." Thus, the approaches taken in Nyhart's BACKGROUND have problems that remain unsolved and teach away from Applicants' claims involving an acoustic (not a sidetone) echo canceller. They teach away from Applicants' claimed embodiments because Nyhart has no concept of utilizing sound output of an entertainment sound adapter of an electronic device to train an acoustic echo canceller of the device in a background of a telecommunications

application. Nyhart strictly relates to telecommunications and, in particular, "In response to noise generated between the dialing of digits, the echo canceller converges on noise to optimize sidetone," (Nyhart, col. 2, ll. 1-4) Nyhart involves "first and second radios and a base station having an echo canceller," (Nyhart, col. 1, ll. 57-60) where the echo canceller is a sidetone echo canceller.

As introduced in the Nyhart ABSTRACT, Nyhart teaches and suggests training "on noise generated by the echo canceller (124) during inter digit dialing." In particular, at col. 3, ll. 11-28, the DSP 124 generates low level white noise in a pseudo random (PN) sequence onto the two wire phone line 126 during inter digit dialing with the result: "The echo canceller 124 is thus trained during the inter digit dialing time before two way communication between the near end and far end users is established." This is not a disclosure or suggestion of Applicants' claimed embodiments using an entertainment sound adapter in an acoustic echo canceller.

Applicants' claims as amended clearly recite distinctions and features that one of ordinary creativity or one using common sense (see *KSR v. Teleflex guidelines re "obviousness"*) in view of Nyhart or the admitted prior art would not obtain without the use of improper hindsight reconstruction. Moreover, Nyhart and the admitted prior art teach away from the recited non-training audio application, for example, an entertainment application playing in the background of a telecommunications application having a first sampling rate and the non-training audio application having a second higher sampling rate.

Nyhart fails to discuss any other audio signal for training than noise generated as low level white noise by DSP 124 (col. 3, ll. 12-28). This is not a disclosure, for example, of "utilizing sound output of an entertainment sound adapter of said electronic device, said entertainment sound adapter output being sampled at a second higher sampling rate than said first sampling rate, said entertainment sound adapter output corresponding to a non-training audio application of said electronic device to train the acoustic echo canceller in a background of said telecommunications application." Nyhart has no entertainment sound adapter and is not an electronic device as recited. There is no concept in Nyhart of a sound output of an entertainment sound adapter for training. Nyhart uses low level white noise. It is not sound output of an entertainment sound adapter of an electronic device. Per claim 2/1, Nyhart, for example, has no "entertainment application" or "program audio" which clearly differentiates from noise. Claim 3/2/1 specifies "streaming audio sound" which is not white noise.

The Examiner suggests at Page 10, regarding claim 10, that "the dialing tones" are used for training; (the Examiner states no grounds of rejection for claim 21). See also, Page 17 of the FOA, regarding dialing tones and an entertainment sound adaptor. Applicants respectfully disagree that dialing tones are used for training. Nyhart, col. 3, explains that the dialed digit is reflected and is used to mute the audio so that the echo canceller can generate *noise* for training: "Referring now to Fig. 3 there is shown a flowchart 300 representing the preferred embodiment describing the training (or convergence) technique used during the dialing sequence . . . As the user dials in the first digit at step 310 the near end piece hears the reflected digit being dialed at step 312. The audio to the near end piece is then muted in step 314. Once the audio to the ear piece is muted *the echo canceller generates noise, preferably low level white noise*, onto the phone line at step 316. The echo canceller converges on the reflected <u>noise</u> signaling step 318. . . allowing it to converge <u>only</u> during <u>inter digit dialing</u> time," (our emphasis added).

The Examiner also suggests that "dial tones" "are streamed for as long as a user pressed the button down" at Page 19, lines 4-5 of the FOA. A tone is not a stream as used in the present specification – by stream, one skilled in the art would understand a digital stream from the specification and context. Stream must be construed in accordance with the Knutson specification. The Examiner fails to construe, for example, "streaming audio sound" of claim 3/2/1 properly.

An advantage of embodiments involving an adaptive filter as an acoustic echo canceller using background training via an entertainment sound adapter is that "background training would not need to operate continuously" as stated at page 13 of the specification (paragraph [0047] of the published application): "idle cycles of the processor can be used to train the echo canceller whenever the speaker is used, whether in video games, playing MP3s, CDs, or other audio files, playing video files, or even during the typical bells and whistles of the PC alerting the user to emails and other warnings." Consequently, no PN noise generation is required as in Nyhart. An entertainment sound adapter of an electronic device is present for other purposes than acoustic echo canceller training. Applicant's amended claims discuss an entertainment sound adapter that is used for non-training and for training of an acoustic echo canceller. The Examiner still does not address this advantage. The Examiner is referred to MPEP 707.07(f): ANSWERING ASSERTED ADVANTAGES. Nyhart is not background training. While Nyhart training occurs during interdigital dialing and so does not need to operate continuously, Nyhart is limited to

operating only during a brief portion of a "telecommunications application" and certainly not in a background of a telecommunications application.

Claim 2/1 as amended reads, for example: "the non-training audio application is an entertainment application," and there is no entertainment application in Nyhart and no "entertainment sound adapter output" or output that includes "program audio". White noise is not entertaining and is not program audio. To the contrary, white noise is annoying.

Claim 3/2/1 as amended refers to streaming audio sound, and Nyhart fails to refer to "streaming audio sound." This is not a dialing tone signal.

Claim 4/1 discusses an entertainment application of said first personal computer, and Nyhart fails to discuss such an application, for example, music and multimedia.

Claim 5/1 relates to matching sample rates as supported at paragraph [0044] to communication sampling rates. The Examiner cannot produce "sample rate conversion" out of thin air from Nyhart – the alleged inherent composition is clearly impermissible hindsight reconstruction of claim 5 from absolutely no disclosure in Nyhart of sample rates, their conversion or matching.

Claim 6/1 relates to "a microphone" and "a speaker" of "said electronic device" of claim 1 which admittedly may be associated with a telecommunication application but must be construed in the context of claim 1. Moreover, an adaptive filter is recited along with paths from the speaker and the microphone to the adaptive filter. Nyhart does not discuss these features in the context of claim 1 including an entertainment sound adapter. Nyhart barely mentions microphone 106 and speaker 108 of handset 104 and so does not relate at all to acoustic echo cancellation.

With respect to claim 7/6/1 and 8/7/6/1, the Examiner states that it would be obvious to balance and manage processor resources. Yet, the Examiner provides no support in Nyhart or any reference to any balancing as recited when both a telecommunications application and a non-training audio application are playing with the latter playing in a background for acoustic echo canceller training. Nyhart only appears to run a communications application. Referring to FIG. 6, the present specification supports a loop 601, 605, 610 which is support for a "processor load, high or low" box (610) and running a canceller continuously (650 when low) depending on the result.

Claim 9/8/7/6/1 is rejected based on another inherency argument. Yet, the Examiner fails to cite to any reference related to, for example, "an adaptive counter to count a number of training calls to the acoustic canceller." The Examiner is again referred to FIG. 6 and the loop 601, 605, 610, 615, 620, 655 where (605) represents "audio application/other sound." If the load is high at (610), incrementing an adaptive counter (615) where if the adaptive counter value is greater than a value (620), the filter is not operated (655). Adaptive counter 854 is introduced at paragraph [0053] and further discussed at paragraphs [0057]-[0062]. Nyhart has no such adaptive counter and fails to discuss the features of claim 9/8/7/6/1.

Claim 10 is an independent claim related to a further embodiment for an acoustic echo canceller involving "a sequence of frequencies" and "an event unrelated to training." Nyhart arguably during digit dialing outputs "a sequence of frequencies" such as so-called touch-tone dialing frequencies (the Examiner may consider white noise or dialing tones as a sequence of frequencies), but Nyhart teaches interdigital training, not during digits. As suggested above, Nyhart does not discuss "utilizing sound output of an entertainment sound adapter of an electronic device . . ." Claim 11/10 defines the event unrelated to training as some event other than outgoing calls involving interdigital dialing so Nyhart does not discuss the recited event. Claims 21 and 22/21 are related and are patentable for the reasons that claims 10 and 11 are patentable.

Claim 12 is an independent claim that relates to an acoustic echo canceller involving "an entertainment sound adapter of an electronic device" and "an adaptive filter adapted to be trained using sound comprising audio output of said entertainment sound adapter" and related features not discussed by Nyhart.

Further claims contain similar features to those already discussed which are not disclosed in Nyhart and are not inherent in Nyhart as suggested by the Examiner. Again, it is respectfully submitted that all such inherency arguments are improper hindsight reconstruction of Applicants' claimed embodiments and requests that some reference be cited which provides a discussion of the alleged inherent component or feature.

Again, at best, Nyhart and the admitted prior art together teach PN sequence noise generation during inter-digit dialing and an adaptive filter for sidetone echo cancellation which has nothing to do with acoustic echo cancellation. Claims 1-27 contain features undisclosed by

the cited and applied prior art such as an entertainment sound adapter. Claim 28 has been added to specify first and second sampling rates of claim 1 (for example, to clarify "higher").

Applicants respectfully request reconsideration of the rejection of claims 1-27 and consideration of new claim 28 and look forward to prompt allowance of the application. The Examiner is urged to contact Thomas Jackson, Registration No. 29,808, located in the District of Columbia to schedule an interview which may include Paul Knutson to answer questions and establish a dialog in this application which may lead to allowance and not impasse. Should the Examiner have any questions on this request, the Examiner is urged to contact the undersigned attorney of record at the telephone number and address given.

Respectfully submitted,
Junbiao Zhang, et al.

By: /Catherine A. Ferguson/_______
Catherine A. Ferguson, Attorney
Reg. No. 40877

Date: __07282009______________

Patent Operations
Thomson Licensing LLC
P. O. Box 5312
Princeton, New Jersey 08543-5312
Telephone: (609) 734-6440

Enclosure: Affidavit under Rule 132 of Benjuan Zhang and attachment